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EXAMINER	
WOZNIAK, JAMES S	

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Please find below and/or attached an Office communication concerning this application or proceeding.

PTO-90C (Rev. 10/03)

Office Action Summary	Application No. 09/927,578	Applicant(s) CLAESSON ET AL.
	Examiner James S. Wozniak	Art Unit 2655

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *08/06/2001*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-57* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-57* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☒ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date 5, 7.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

Detailed Action

Information Disclosure Statement

1. The electronic information disclosure statement (EIDS) submitted on 1/23/04 has been considered by the examiner.

Specification

2. The title of the invention is not descriptive, with respect to copending applications 09/669,069 and 10/214,944, which have the same title. A new title is required that is clearly indicative of the invention to which the claims are directed and differentiates the present invention from 09/669,069 and 10/214,944.

Double Patenting

3. The nonstatutory double patenting rejection is based on a judicially created doctrine grounded in public policy (a policy reflected in the statute) so as to prevent the unjustified or improper timewise extension of the "right to exclude" granted by a patent and to prevent possible harassment by multiple assignees. See *In re Goodman*, 11 F.3d 1046, 29 USPQ2d 2010 (Fed. Cir. 1993); *In re Longi*, 759 F.2d 887, 225 USPQ 645 (Fed. Cir. 1985); *In re Van Ornum*, 686 F.2d 937, 214 USPQ 761 (CCPA 1982); *In re Vogel*, 422 F.2d 438, 164 USPQ 619 (CCPA 1970);and, *In re Thorington,* 418 F.2d 528, 163 USPQ 644 (CCPA 1969).

A timely filed terminal disclaimer in compliance with 37 CFR 1.321(c) may be used to overcome an actual or provisional rejection based on a nonstatutory double patenting ground provided the conflicting application or patent is shown to be commonly owned with this application. See 37 CFR 1.130(b).

Effective January 1, 1994, a registered attorney or agent of record may sign a terminal disclaimer. A terminal disclaimer signed by the assignee must fully comply with 37 CFR 3.73(b).

4. **Claims 1-57** are provisionally rejected under the judicially created doctrine of obviousness-type double patenting as being unpatentable over claims 1-19 of copending Application No. 09/669,069. Although the conflicting claims are not identical, they are not patentably distinct from each other because it would have been obvious to one of ordinary skill in the art, at the time of invention, to implement the digital audio clarification method of the present invention with digital audio transmission platforms and receiving devices using the well-known platforms described in Claims 26-30, 32-36, well-known audio devices as recited in Claims 39, 40 and the well-known and commonly used MP3 coding format as recited in Claim 46 to improve audio signal quality since all of the applications are well-known utilizations of digital audio. Also, with respect to Claims 9-11 and in view of Mead et al (U.S. Patent: 6,044,162), it would have been obvious to utilize future samples to reduce an initial gain and gain-limit subbands that exceed a threshold according to an acoustic intensity threshold while appropriately amplifying a first set of samples that may require amplification in order to increase

audio clarity without over-amplifying signal segments that may already possess a perceptible audio level, according to the below art rejection of Claims 9-11.

This is a provisional obviousness-type double patenting rejection because the conflicting claims have not in fact been patented.

Claim Rejections - 35 USC § 112

5. The following is a quotation of the first paragraph of 35 U.S.C. 112:

> The specification shall contain a written description of the invention, and of the manner and process of making and using it, in such full, clear, concise, and exact terms as to enable any person skilled in the art to which it pertains, or with which it is most nearly connected, to make and use the same and shall set forth the best mode contemplated by the inventor of carrying out his invention.

6. **Claims 26-30 and 32-36** are rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the enablement requirement. The claims contain subject matter that was not described in the specification in such a way as to enable one skilled in the art to which it pertains, or with which it is most nearly connected, to make and/or use the invention.

Specifically: the server platform in a WAN, digital radio transmission platform, cellular communication transmission platform, cable television transmission platform, or satellite television transmission platform can be implemented in a number of ways using a variety of different elements. These elements, that would comprise any of these platforms to provide functionality, are not disclosed in the specification and thus, it would be unclear to one of ordinary skill in the art exactly how these specific platforms, which utilize the present invention, could be implemented.

Furthermore: the client platform in a WAN, digital radio receiver, portable cellular communication device, cable television decoder, or satellite television decoder can also be implemented in a number of way using a variety of different elements. Thus the implementation of these specific claimed devices would be unclear to one of ordinary skill in the art for the same reasons as noted above.

Claim Rejections - 35 USC § 103

7. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

8. **Claims 1-6, 20, 21, and 25-40** are rejected under 35 U.S.C. 103(a) as being unpatentable over Lindemann et al *(U.S. Patent 6,097,824 of WO 98/56210)*.

With respect to **Claim 1**, Lindemann discloses:

First instructions for separating the original sampled signal into a plurality of signal components each corresponding to one of a plurality of frequency bands *(filtering an audio signal into a plurality of frequency bands, Col. 3, Lines 55-56)*;

Second instructions for independently and dynamically controlling a dynamic range associated with each one of the plurality of signal components *(dynamic range compression gain calculation and application, Col. 5, Lines 40-46)*;

Third instructions for modifying at least one signal level associated with the plurality of signal components *(multiplying a band by a respective gain, Col. 5, Lines 45-46)*; and

Fourth instructions for combining the signal components into a processed sampled signal *(summing scaled bands to generate an output signal, Col. 5, Lines 46-47)*.

Lindemann does not teach method implementation using instructions stored on a computer readable medium, however, it would have been obvious to one of ordinary skill in the art, at the time of invention, to store the instructions for multi-band audio compression on a computer readable medium to increase method compatibility and usability by providing a means for method use with multiple computer systems.

With respect to **Claim 2**, Lindemann teaches filtering an audio signal into a plurality of overlapping frequency bands (Col. 5, Lines 32-33). Lindemann does not specifically disclose separating an audio signal into one of 3, 4, or 5 bands; however, it would have been obvious to one of ordinary skill in the art, at the time of invention, to divide the audio signal into one of 3, 4, or 5 bands since those specific band amounts fall within the scope of the plurality of overlapping frequency bands as disclosed by Lindemann as a matter of design choice since the applicant has not disclosed that dividing an audio signal into a specific 3, 4, or 5 frequency bands solves any stated problem.

With respect to **Claim 3**, Lindemann discloses:

The second instructions effect nonlinear control of a gain factor associated with each of the signal components *(compression gain calculation for each band, based upon a power signal and according to a predetermined function, Col. 5, Lines 40-46)*.

With respect to **Claim 4**, Lindemann recites:

The second instructions control the dynamic range associated with each of the signal components by applying a gain factor to each sample of each of the signal components, the gain factor being dynamically adjusted *(determining a dynamic range compression gain based for each frequency band, based upon respective band power signals, Col. 5, Lines 40-46)*.

With respect to **Claim 5**, Lindemann teaches the use of an analog-to-digital converter for obtaining a digital audio signal, Col. 5, Lines 17-18, which is further divided into frequency bands, Col. 5, Lines 32-33. Lindemann does not specifically teach that a gain factor is adjusted every first number of samples, however, it would have been obvious to one of ordinary skill in the art, at the time of invention, that since the sampled signal is divided into frequency bands and a gain factor is calculated for each band based upon an associated estimated power signal, as applied to Claim 1, each band would contain a specific amount of samples, and thus, a gain factor would be calculated for every specific number of samples comprising a frequency band.

With respect to **Claim 6**, Lindemann teaches the use of an analog-to-digital converter for obtaining a digital audio signal, Col. 5, Lines 17-18, which is further divided into frequency bands (comprising a specific number of samples), Col. 5, Lines 32-33, each having an associated calculated dynamic range compression gain, as applied to Claim 5. Lindemann does not specifically teach that a gain factor is adjusted every 64 samples, however, it would have been obvious matter of design choice to calculate a dynamic range compression gain every 64 samples of an audio signal, since the applicant has not disclosed that calculating a gain every 64 samples solves any stated problem or is for any particular purpose. Also, adjusting the gain factor every 64 samples would allow for sufficient level adjustment increments over an entire audio signal for perceptibility enhancement.

With respect to **Claims 20 and 21**, Lindemann teaches the filtering an audio signal into a plurality of overlapping frequency bands (Col. 5, Lines 32-33), but does not specifically suggest dividing an audio signal into two or three bands (two way or three way crossover). However, the examiner takes official notice that it is well known in the art to divide an audio signal into any number of frequency bands as an obvious matter of design choice based on the user's desired level of accuracy, with a more detailed level of adjustment corresponding to increasing band numbers. Thus, it would have been obvious that an audio signal could be divided into any number of frequency bands, including two or three, based upon the user's desired level of sound level adjustment accuracy. Furthermore, the background of the invention specifies that multi-band compressors commonly consist of two or three frequency bands *(Col. 1, Lines 39-40)*.

With respect to **Claim 25**, Lindemann teaches the dynamic range audio compressor as applied to Claim 1. Lindemann does not specifically suggest device and method use with a digital audio transmission system, however, since Lindemann discloses the modification of digital audio data and the examiner takes official notice that it is well known in the art to transmit digital audio from a server in an application such as streaming audio for Internet radio, it would have been obvious to one of ordinary skill in the art, at the time of invention, to implement the audio processing method taught by Lindemann in a well-known application of digital audio usage in a digital audio transmission system to improve audio signal quality before transmission.

With respect to **Claims 26-30**, Lindemann teaches the dynamic range audio compressor as applied to Claim 1. Lindemann does not specifically suggest device and method use with a digital audio transmission system specifically featuring: a server platform in a WAN, a digital radio transmission platform, a cellular communication transmission platform, a cable television

transmission platform, or a satellite television transmission platform, however, since Lindemann discloses the modification of digital audio data and the examiner takes official notice that the aforementioned platforms are well known platforms for the transmission of digital audio data, it would have been obvious to one of ordinary skill in the art, at the time of invention, to implement the audio processing method taught by Lindemann in the aforementioned and well-known applications of digital audio transmission to improve audio signal quality before transmission.

With respect to **Claim 31**, Lindemann teaches the dynamic range audio compressor as applied to Claim 1. Lindemann does not specifically suggest device and method use with a digital audio receiving system, however, since Lindemann discloses the modification of digital audio data and the examiner takes official notice that it is well known in the art to receive digital audio from a server, in an application such as streaming audio for Internet radio using a computer, it would have been obvious to one of ordinary skill in the art, at the time of invention, to implement the audio processing method taught by Lindemann in a well-known application of digital audio usage in a digital audio receiving system to improve audio signal quality upon reception.

With respect to **Claims 32-36**, Lindemann teaches the dynamic range audio compressor as applied to Claim 1. Lindemann does not specifically suggest device and method use with a digital audio receiving system specifically featuring: a client platform in a WAN, a digital radio receiver, a portable cellular communication device, a cable television decoder, or a satellite television decoder, however, since Lindemann discloses the modification of digital audio data and the examiner takes official notice that the aforementioned devices are well known devices

for receiving digital audio data, it would have been obvious to one of ordinary skill in the art, at the time of invention, to implement the audio processing method taught by Lindemann in the aforementioned and well-known devices for digital audio reception to improve audio signal quality upon reception.

With respect to **Claims 37-40**, Lindemann teaches the dynamic range audio compressor as applied to Claim 1. Lindemann does not specifically suggest device and method use with portable digital audio devices such as CD and MP3 players; however, since Lindemann discloses the modification of digital audio data and the examiner takes official notice that portable audio devices such as MP3 or CD players are well-known devices utilizing digital audio data, it would have been obvious to one of ordinary skill in the art, at the time of invention, to implement the audio processing method taught by Lindemann in a well-known application of digital audio usage in a portable audio device such as an MP3 or CD player to improve audio signal quality.

9. **Claims 7-9, 12-19, 22-24, and 41-57** are rejected under 35 U.S.C. 103(a) as being unpatentable over Lindemann et al in view of Allen et al *(U.S. Patent: 5,524,148)*.

With respect to **Claim 7**, Lindemann teaches the dynamic range audio compressor capable of calculating a dynamic range compression gain for each frequency band based upon an associated estimated power signal, as applied to Claim 4. Lindemann does not specifically suggest adjusting a gain factor with reference to a threshold level, however, Allen discloses:

The gain factor for each of the signal components is dynamically adjusted with reference to a threshold level to which each sample of each of the signal components is compared

(adjusting a gain factor based upon comparison to a "breakpoint" threshold, Col. 8, Lines 12-27).

Lindemann and Allen are analogous art because they are from a similar field of endeavor in audio signal processing. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to combine the method of adjusting a gain factor based upon comparison to a signal level threshold as taught by Allen with the dynamic range audio compressor capable of calculating a dynamic range compression gain for each frequency band based upon an associated estimated power signal as taught by Lindemann in order to differentiate audio data from noise and calculate the necessary gain required to produce a perceptible audio signal for further applying a required gain amount to provide increased audio clarity without over amplifying noise *(Allen, Col. 7, Lines 58-60)*. Therefore, it would have been obvious to combine Allen with Lindemann for the benefit of obtaining increased audio clarity by applying a required gain factor obtained from a threshold comparison, to obtain the invention as specified in Claim 7.

With respect to **Claim 8**, Lindemann teaches the dynamic range audio compressor capable of calculating a dynamic range compression gain for each frequency band based upon an associated estimated power signal, as applied to Claim 4. Lindemann does not teach adjusting a gain factor upward using a release rate and downward using an attack rate, however, Allen discloses:

The gain factor is adjusted upward using a release rate parameter where each sample is below the threshold level, and downward using an attack rate parameter where each sample is above the threshold level *(threshold comparison, Col. 8, Lines 12-27, an attack time for reducing compressor gain, and a release time for increasing compressor gain, Col. 8, Lines 47-57).*

Lindemann and Allen are analogous art because they are from a similar field of endeavor in audio signal processing. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to combine the use of an attack time for reducing compressor gain and a release time for increasing gain as taught by Allen with the dynamic range audio compressor capable of calculating a dynamic range compression gain for each frequency band based upon an associated estimated power signal as taught by Lindemann to implement a means (using attack and release times) for instantaneously reducing or increasing audio signal levels to provide increased perceptibility for fast-changing audio and prevent loud noises from being over-amplified. Therefore, it would have been obvious to combine Allen with Lindemann for the benefit of obtaining increased audio clarity, while preventing the over-amplification of noise signals by utilizing attack and release times, to obtain the invention as specified in Claim 8.

With respect to **Claim 9**, Lindemann teaches the dynamic range audio compressor as applied to Claim 1. Lindemann does not teach limiting the gain with respect to future samples, however Allen recites:

The third instructions limit the at least one signal level with reference to a first number of future samples (peak detector that controls attack time to limit the gain, as applied to Claim 8, and peak detector consideration of a future input sample ($y(n)=x(n)$ if $x(n)>y(n-1)$) to determine an attack time, Col. 8, Lines 47-64).

Lindemann and Allen are analogous art because they are from a similar field of endeavor in audio signal processing. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to combine the use of future samples in determining an attack time to limit gain as taught by Allen with the dynamic range audio compressor as taught by

Lindemann to prevent a sudden and prolonged noise signal from being over-amplified in the case that the noise signal should be situated at the start of an audio signal by considering future sample signal levels for further calculation of an attack time to limit gain. Therefore, it would have been obvious to combine Allen with Lindemann for the benefit of preventing the over-amplification of sudden and prolonged noise signals by considering future sample signal levels for further calculation of an attack time to limit gain, to obtain the invention as specified in Claim 9.

With respect to **Claims 12 and 13**, Lindemann teaches the dynamic range audio compressor as applied to Claim 1. Lindemann does not teach modifying a gain factor using an attack rate, however, Allen discloses:

The third instructions implement an independent negative attack time limiter for application to each of the plurality of signal components from a sampled signal analog-to-digital converter for obtaining a digital audio signal, Col. 5, Lines 17-18, *which is further divided into frequency bands, peak detector that controls an attack time in order to modify and limit gain for each frequency band, Col. 8, Lines 48-56).*

Lindemann and Allen are analogous art because they are from a similar field of endeavor in audio signal processing. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to combine an attack time implemented by a peak detector to modify and limit gain within frequency bands of a sampled signal as taught by Allen with the dynamic range audio compressor to implement a means (using an attack time) for instantaneously reducing audio signal levels to prevent loud audio from being over-amplified. Also, it would have been obvious to one of ordinary skill in the art, at the time of invention, to separately

implement an attack time calculation device in order to individually calculate an attack time in the situation that an audio signal should exceed a threshold in order to provide for more efficient audio processing. Therefore, it would have been obvious to combine Allen with Lindemann for the benefit of preventing the over-amplification of loud audio signals by utilizing an attack time applied to the associated gains of frequency bands of a sampled signal, to obtain the invention as specified in Claims 12 and 13.

With respect to **Claims 14 and 15**, Lindemann teaches the dynamic range audio compressor as applied to Claim 1. Lindemann does not teach an additional step of applying preset gain factors to multiple frequency bands, however Allen discloses:

Fifth instructions for applying at least one preset gain factor to at least one of the processed sampled signal and the plurality of signal components, wherein the at least one preset gain factor comprises a plurality of preset gain factors, each preset gain factor corresponding to one of the plurality of signal components (applying a predetermined low level gain, GL and predetermined high level gain, GH to subbands with appropriate energy levels, Col. 8, Lines 1-4).

Lindemann and Allen are analogous art because they are from a similar field of endeavor in audio signal processing. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to combine the application of multiple preset gains corresponding to various energy levels within specific subbands as taught by Allen with the dynamic range audio compressor as taught by Lindemann to provide for more efficient audio processing by bypassing any type of necessary gain modification if audio levels within subbands can be sufficiently corrected, with respect to perceptibility, using preset gains. Therefore, it would have been

obvious to combine Allen with Lindemann for the benefit of providing more efficient audio processing utilizing preset gains to bypass unnecessary gain modification when appropriate, to obtain the invention as specified in Claims 14 and 15.

With respect to **Claim 16**, Lindemann teaches the dynamic range audio compressor as applied to Claim 1. Lindemann does not teach preset gain factors corresponding to multiple frequency bands, however Allen discloses:

Multiple ones of the plurality of preset gain factors correspond to each of the plurality of signal components *(high level predetermined gain, GH, corresponding to high energy frequency bands, and low level predetermined gain, GL, corresponding to low energy frequency bands, Col. 8, Lines 24-27).*

Lindemann and Allen are analogous art because they are from a similar field of endeavor in audio signal processing. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to combine the use of multiple predetermined gains corresponding to the energy levels of specific frequency bands as taught by Allen with the dynamic range audio compressor as taught by Lindemann to provide for more efficient audio processing by bypassing any type of necessary gain modification if audio levels can be sufficiently corrected, with respect to perceptibility, using preset gains corresponding to energy levels within specific frequency bands. Therefore, it would have been obvious to combine Allen with Lindemann for the benefit of providing more efficient audio processing utilizing preset gains corresponding to signal levels within a subband to bypass unnecessary gain modification when appropriate, to obtain the invention as specified in Claim 16.

With respect to **Claim 17**, Lindemann in view of Allen teaches the dynamic range audio compressor utilizing preset gains to alter signal levels within various frequency bands, as applied to Claim 16. Lindemann in view of Allen does not specifically suggest the use of an inverse gain factor, however, it would have been obvious to one of ordinary skill in the art, at the time of invention, to utilize an inverse gain factor to correct for an unnecessarily applied gain (that over-amplifies an audio signal portion), by undoing the gain factor with its inverse. Therefore, to correct an inappropriate gain factor, it would have been obvious to implement an inverse gain factor to cancel out the frequency band gain.

With respect to **Claim 18**, Lindemann in view of Allen teaches the dynamic range audio compressor utilizing preset gains to alter signal levels within various frequency bands, as applied to Claim 16. Lindemann in view of Allen does not specifically suggest applying a gain and then an inverse gain before and after a step of modifying a frequency band, however, it would have been obvious to one of ordinary skill in the art, at the time of invention, to cancel out a gain by applying an inverse gain to prevent undesirable audio amplification in the case that, after applying or modifying a gain, it was determined that the gain was inappropriately applied or modified and thus unnecessary (i.e., over-amplifying an audio signal portion). Therefore, to correct an inappropriate gain factor, it would have been obvious to implement an inverse gain factor, after the application of a gain factor or modified gain factor to alter a signal level, to cancel out the frequency band gain to prevent undesirable audio amplification.

With respect to **Claim 19**, Lindemann teaches the dynamic range audio compressor as applied to Claim 1. Lindemann does not teach a first present gain factor applied to various frequency bands, however, Allen recites:

The at least one preset gain factor comprises a first preset gain factor for applying to the processed sampled signal (*high level predetermined gain, GH, corresponding to high energy frequency bands or low level predetermined gain, GL, corresponding to low energy frequency bands applied to a frequency band before modification, Col. 8, Lines 24-27)*.

Lindemann and Allen are analogous art because they are from a similar field of endeavor in audio signal processing. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to combine the use of a high or low level predetermined gain applied to a frequency band before modification as taught by Allen with the dynamic range audio compressor taught by Lindemann in order to provide for more efficient audio processing by bypassing any type of necessary gain modification if audio levels within subbands can be sufficiently corrected, with respect to perceptibility, using preset gains. Therefore, it would have been obvious to combine Allen with Lindemann for the benefit of providing more efficient audio processing utilizing preset gains to bypass unnecessary gain modification when appropriate, to obtain the invention as specified in Claim 19.

With respect to **Claim 22**, Lindemann in view of Allen teaches the dynamic range audio compressor utilizing preset gains (further altered using attack or release times) to modify signal levels within various frequency bands and an inverse gain to cancel out an unnecessary gain that may produce undesirable audio effects, as applied to Claims 8, 13, 15, 18, and 20.

Lindemann in view of Allen does not specifically suggest the use of four two-way crossover blocks, five AGC blocks, and five NATL blocks, however, as noted above with respect to Claims 20 and 21, the examiner takes official notice that it is well known in the art to utilize any required number or type of filters (two-way crossovers- high and low pass filters) in stages,

to divide and subdivide an audio signal into subbands, as a matter of design choice based on the user's desired level of accuracy, with a more detailed level of adjustment corresponding to increasing band numbers. Also, it would have been obvious to one of ordinary skill in the art that utilizing 4 two-way crossovers to divide and further subdivide an audio signal into subbands, would create 5 subbands, that would each require an associated gain and attack time, thus necessitating 5 AGC and NATL blocks. Therefore, as a matter of design choice, it would have been obvious to one of ordinary skill in the art, at the time of invention, to utilize any number or type of filters (4 two-way crossovers) in stages to divide and subdivide an audio signal into subbands (specifically, 5 in this case), based upon the user's desired level of sound level adjustment accuracy.

With respect to **Claim 23**, Lindemann in view of Allen teaches the dynamic range audio compressor utilizing preset gains (further altered using attack or release times) to modify signal levels within various frequency bands and an inverse gain to cancel out an unnecessary gain that may produce undesirable audio effects, as applied to Claims 8, 13, 15, 18, and 20.

Lindemann in view of Allen does not specifically suggest the use of two three-way crossover blocks, five AGC blocks, and five NATL blocks, however, as noted above with respect to Claims 20 and 21, the examiner takes official notice that it is well known in the art to utilize any required number or type of filters (three-way crossovers- high, bandpass, and low pass filters) in stages, to divide and subdivide an audio signal into subbands, as a matter of design choice based on the user's desired level of accuracy, with a more detailed level of adjustment corresponding to increasing band numbers. Also, it would have been obvious to one of ordinary skill in the art that utilizing 2 three-way crossovers to divide (creating 3 bands) and further

subdivide (creating 3 bands from one of the initially divided bands for a total of 5) an audio signal into subbands, would create 5 subbands, that would each require an associated gain and attack time, thus necessitating 5 AGC and NATL blocks. Therefore, as a matter of design choice, it would have been obvious to one of ordinary skill in the art, at the time of invention, to utilize any number or type of filters (2 three-way crossovers) in stages to divide and subdivide an audio signal into subbands (specifically, 5 in this case), based upon the user's desired level of sound level adjustment accuracy.

With respect to **Claim 24**, Lindemann in view of Allen teaches the dynamic range audio compressor utilizing preset gains (further altered using attack or release times) to modify signal levels within various frequency bands and an inverse gain to cancel out an unnecessary gain that may produce undesirable audio effects, as applied to Claims 8, 13, 15, 18, and 20.

Lindemann in view of Allen does not specifically suggest the use of 1 two-way and 1 three-way crossover blocks, four AGC blocks, and four NATL blocks, however, as noted above with respect to Claims 20 and 21, the examiner takes official notice that it is well known in the art to utilize any required number or type of filters (two-way crossover- high and low pass filters and three-way crossover- high, bandpass, and low pass filters) in stages, to divide and subdivide an audio signal into subbands, as a matter of design choice based on the user's desired level of accuracy, with a more detailed level of adjustment corresponding to increasing band numbers. Also, it would have been obvious to one of ordinary skill in the art that utilizing 1 two-way and 1 three-way crossover to divide (creating 2 bands) and further subdivide (creating 3 bands from one of the initially divided bands for a total of 4) an audio signal into subbands, would create 4 subbands, that would each require an associated gain and attack time, thus necessitating 4 AGC

and NATL blocks. Therefore, as a matter of design choice, it would have been obvious to one of ordinary skill in the art, at the time of invention, to utilize any number or type of filters (1 two-way and 1 three-way crossover) in stages to divide and subdivide an audio signal into subbands (specifically, 4 in this case), based upon the user's desired level of sound level adjustment accuracy.

Claims 41, 52, and 53 contain subject matter similar to Claims 1 and 8, and thus, are rejected for the same reasons.

Claim 42 contains subject matter similar to Claims 26 and 32, and thus is rejected for the same reasons.

Claim 43 contains subject matter similar to Claim 26, and thus is rejected for the same reasons.

Also, as noted above with respect to Claim 26, the examiner takes official notice that it would have been obvious to modify an audio signal at a server, prior to transmission, in order to provide a high quality audio signal to a receiving device (with limited processing means).

Claim 44 contains subject matter similar to Claim 32, and thus is rejected for the same reasons.

Also, as noted above with respect to Claim 32, the examiner takes official notice that it would have been obvious to modify an audio signal at a receiving device, upon reception, in order to provide a high quality audio signal to a listener (when a server features limited processing means).

With respect to **Claims 45 and 46**, Lindemann in view of Allen teaches the dynamic range audio compressor as applied to Claim 41. Lindemann in view of Allen does not

specifically suggest encoding the compressed audio signal in an MP3 format, however, the examiner takes official notice that it is well known in the art to encode an audio signal into a compressed audio format such as MP3, in order to conserve bandwidth, in a communication medium such as the Internet, and enable quicker access to the files due to decreased file size.

Claims 47-50 contain subject matter similar to Claims 26, 28, 32, 34, 39, 40, and 46, and thus, are rejected for the same reasons.

Claim 51 contains subject matter similar to Claim 40 (well-known MP3 player that utilizes a memory card for storing MP3 audio files), and thus is rejected for the same reasons.

Claim 54 contains subject matter similar to Claim 22, and thus, is rejected for the same reasons.

Claim 55 contains subject matter similar to Claim 23, and thus, is rejected for the same reasons.

Claim 56 contains subject matter similar to Claim 24, and thus, is rejected for the same reasons.

With respect to **Claim 57**, Lindemann in view of Allen teaches the dynamic range audio compressor utilizing preset gains (further altered using attack or release times) to modify signal levels within various frequency bands and an inverse gain to cancel out an unnecessary gain that may produce undesirable audio effects upon combination and playback, as applied to Claims 8, 13, 15, 18, and 20.

Lindemann in view of Allen does not specifically suggest the use of two two-way crossover blocks, three AGC blocks, three NATL blocks, three preset gain blocks, and three inverse gain blocks, however, as noted above with respect to Claims 20 and 21, the examiner

takes official notice that it is well known in the art to utilize any required number or type of filters (two-way crossovers- high and low pass filters) in stages, to divide and subdivide an audio signal into subbands, as a matter of design choice based on the user's desired level of accuracy, with a more detailed level of adjustment corresponding to increasing band numbers. Also, it would have been obvious to one of ordinary skill in the art that utilizing 2 two-way crossovers to divide (creating 2 subbands) and further subdivide (creating 2 subbands from one of the originally divided subbands for a total of 3) an audio signal into subbands, would create 3 subbands, that would each require an associated gain and attack time, thus necessitating 3 AGC, NATL, gain, and inverse gain blocks. Therefore, as a matter of design choice, it would have been obvious to one of ordinary skill in the art, at the time of invention, to utilize any number or type of filters (2 two-way crossovers) in stages to divide and subdivide an audio signal into subbands (specifically, 3 in this case), based upon the user's desired level of sound level adjustment accuracy.

10. **Claims 10 and 11** are rejected under 35 U.S.C. 103(a) as being unpatentable over Lindemann in view of Allen et al and further in view of Mead et al *(U.S. Patent: 6,044,162)*.

With respect to **Claims 10 and 11,** Lindemann in view of Allen teaches the dynamic range audio compressor as applied to Claim 9. Lindemann in view of Allen does not teach modifying the gain with respect to future samples and if a future sample exceeds a threshold, decreasing the gain in response, however Mead recites:

The gain factor is decreased *(automatic gain control featuring a digital bandpass filter having its output connected to its input, Col. 3, Lines 35-39, thus considering future samples*

when adjusting gain) when application of the gain factor to the at least one future sample results in the at least one future sample exceeding a threshold *(maximum acoustic intensity threshold that corresponds to an upper comfort level of sound, Col. 10, Lines 40-45)*.

Although Mead does not specifically suggest decreasing a gain factor in the case of exceeding a threshold, it would have been obvious to one of ordinary skill in the art, at the time of invention, to decrease the gain factor after it has been applied to the first number of current samples, so that those future samples that exceed the threshold can be gain limited according to the acoustic intensity threshold taught by Mead while the first set of samples that may require amplification can still be appropriately amplified in order to increase audio clarity without over-amplifying signal segments that may already possess a perceptible audio level.

Lindemann, Allen, and Mead are analogous art because they are from a similar field of endeavor in audio signal processing. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to combine the use of future samples in limiting and modifying gain of a present and future segment, including the case wherein gain reduction is applied to future samples that exceed a threshold, while still maintaining amplification of a lower level audio in a present segment as taught by Mead with the dynamic range audio compressor as taught by Lindemann in view of Allen to prevent a sudden and prolonged noise signal from being over-amplified in the case that the noise signal should be situated at the start of an audio signal by considering future sample signal levels to reduce an initial gain and gain limit subbands that exceed a threshold according to an acoustic intensity threshold while appropriately amplifying a first set of samples that may require amplification in order to increase audio clarity without over-amplifying signal segments that may already possess a perceptible audio level.

Therefore, it would have been obvious to combine Mead with Lindemann in view of Allen for the benefit of increasing audio clarity without over-amplifying signal segments that may already possess a perceptible audio level, to obtain the invention as specified in Claims 10 and 11.

Conclusion

11. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure:

- Dillon *(U.S. Patent: 4,803,732)*- teaches a method for the amplification of an audio signal in a hearing aid that features a plurality of bandpass filters for dividing a signal into subbands, amplification means, and signal combining means.
- Jones et al *(U.S. Patent: 5,724,340)*- discloses a gain factor generator for an audio signal that uses a non-linear sliding filter to consider future samples in a delay buffer for gain calculation.
- Schmidt *(U.S. Patent: 5,832,444)*- teaches a dynamic range compression method utilizing attack and release rates to affect control of frequency subband gains.
- Smyth et al *(U.S. Patent: 5,956,674)*- discloses a subband audio coder utilizing gain prediction.

12. Any inquiry concerning this communication or earlier communications from the examiner should be directed to James S. Wozniak whose telephone number is (703) 305-8669

and email is James.Wozniak@uspto.gov. The examiner can normally be reached on Mondays-Fridays, 8:30-4:30.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Doris To can be reached at (703) 305-4827. The fax/phone number for the Technology Center 2600 where this application is assigned is (703) 872-9306.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the technology center receptionist whose telephone number is (703) 306-0377.

James S. Wozniak
8/23/04

W. R. YOUNG
PRIMARY EXAMINER
Acting SPE AU2655